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**Original: English****Question(s):** 14/12**STUDY GROUP 12 – CONTRIBUTION 53****Source:** Chemnitz University of Technology, Germany**Title:** Improvement of network-based QoE estimation for TCP based streaming services**1 Introduction:**

Progressive download video services, such as YouTube and podcasts, are responsible for a major part of the transmitted data volume in the Internet and it is expected, that they will also strongly affect mobile networks. Streaming video quality mainly depends on the sustainable throughput achieved during transmission. To ensure acceptable video quality in mobile networks (with limited capacity resources) the perceived quality by the customer (QoE) needs to be monitored by estimation. For that, the streaming video quality needs to be measured and monitored permanently. For TCP based progressive download we propose to extract the the video timestamps which are encoded within the payload of the TCP segments by decoding the video within the payload. The actual estimation is then done by play out buffer fill level calculations based on the TCP segment timestamp and their internal play out timestamp. The perceived quality for the user is derived from the number and duration of video stalls.

Algorithms for decoding Flash Video, MP4 and WebM Video have already been implemented. After deriving the play out time it is compared to the timestamp of the respective TCP segment. The result of this comparison is an estimate of the fill level of the play out buffer in terms of play out time within the client. This estimation is done without access to the end device.

The same measurement procedure can be applied for any TCP based progressive download Internet service. Video was simply taken as an example because of its current large share in traffic volume in operator networks.

**2 Proposal:**

The proposal is focusing on progressive download video, but is applicable for any TCP based progressive download Internet service.

As improvement of the standard G.1201 which has been released last year we propose the following enhancements to improve processing speed, estimation accuracy and perform failure allocation.

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## **2.1 Increasing processing Speed of QoE estimation**

For all methods the following steps have to be carried out in advance: (1) the video flow needs to be identified within the traffic mix, and (2) the TCP segment information and the TCP payloads of the video flow have to be extracted. Whether it is possible to use speed enhanced processing methods depends on the container format. At the moment the speed up has been proven for WebM Video and MP4 container formats.

### **2.1.1 Estimation Method**

Most accurate estimation results can be achieved by analysing every single TCP segment belonging to a progressive download flow. However, this causes high processing load and might not be really required if the video payload is known by its chunk structure and its respective play out contribution. Therefore, better processing performance can be achieved without a loss in estimate precision. The idea is to decode only the header of the streamed data in full detail. The collected video information in the header includes the respective size and duration information of all subparts (chunks). The estimation algorithm calculates the fill-level based on those chunk sizes and simply the observed amount of data streamed. This calculation yields the number and duration of re-buffering events. Note, that there is a trade-off between processing speed-up and accuracy.

### **2.1.2 Combination of the two Methods**

The second variant of the algorithm tries to combine the gained speed-up of the estimation method with the accuracy of the exact algorithm (full decoding variant). This is realized by dynamically adopting the processing mode (exact mode/estimated mode) to the experienced throughput. The adoption is based on a single threshold value of the buffer fill-level: the exact method is used only if the buffer fill-level is below the threshold.

## **2.2 Increasing Accuracy by using Acknowledgements**

The estimation accuracy of the algorithms can be further improved by using the timestamp of the TCP acknowledgements (ACKs) instead of the timestamp of the TCP data segments in the buffer fill-level calculation. With this improvement it is additionally possible to determine whether a packet reaches the client device after passing through the measurement point. Out of the comparison between observed TCP segments and ACKs the algorithm can also determine the location of the congestion whether it lays in front of or behind the point of measurement in the network. The limitation of this improvement is the required route pinning for the traffic to ensure, that the monitoring entity can actually observe the downstream TCP segments and their corresponding ACKs. However, an split architecture can be deployed as long as the timestamping is synchronized between the downstream and upstream monitoring component.

## **2.3 Failure allocation**

The aim is to determine, whether the streaming impairment is incurred before the point of measurement or after. Working out this indication is done by means of a correlation of segments monitored and associated ACK observed in the opposite direction. If the downstream segments are coming in slower than required, the cause of the quality deterioration must be before the point of measurement. Otherwise, if the respective ACKs are missing, the path after the point of measurement towards the client machine seems to lose packets.

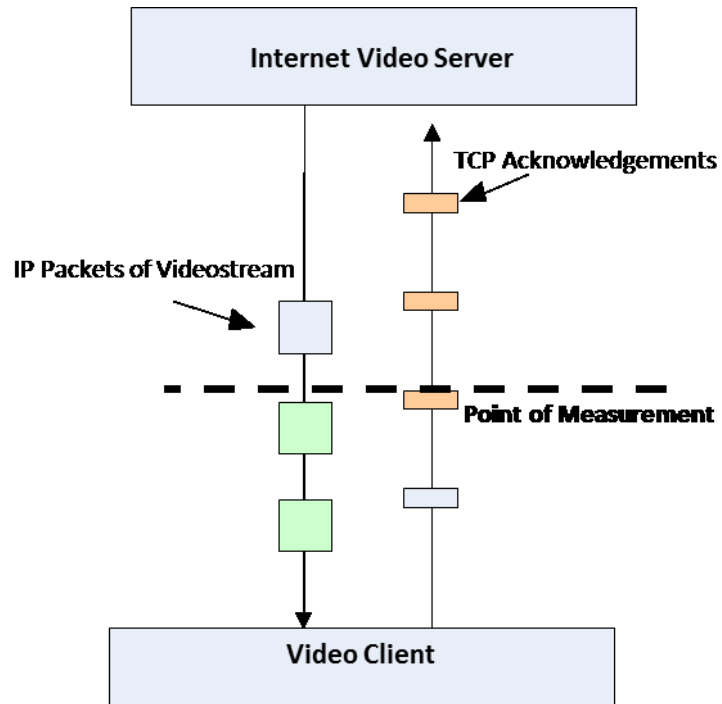


Figure 1: TCP Flow chart

### 3 Acknowledgement

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